



TA100/200

User Manual



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WEEE Warning



In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

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About This Guide

Thanks for choosing Yeastar TA100/200 Analog Telephone Adapter. Yeastar TA100/200 provides 1 or 2 analog interfaces for residential and small business users to convert existing analog equipment to IP-based networks cost effectively. Yeastar TA100/200 is ideal for small business to achieve quick and easy connection in various network environments.

Audience

This manual will help you learn how to operate and manage your TA100/200 Analog Telephone Adapter. In this guide, we describe every detail on the functionality and configuration of TA100/200. We begin by assuming that you are interested in TA100/200 and familiar with networking and other IT disciplines.

Safety when working with electricity



- Do not open the device when the device is powered on.
- Do not work on the device, connect or disconnect cables when lightning strikes.

Feature Highlights

- Connect up to 2 analog phones/faxes
- Miniature in design
- Fully compliant with SIP standard
- High-quality voice call
- Rich subscribe calling features
- FTP, TFTP, HTTP Auto Provision
- Powered by USB interface
- Easy Web-based configuration

Learn more about Yeastar TA100/200 here:

<http://www.yeastar.com/Products.html/Analog-Telephone-Adapter>

Installation

This chapter provides the following sections:

- [TA100/200 Packing List](#)
- [Specifications and Operating Environment](#)
- [Connecting Your TA100/200](#)

TA100/200 Packing List

Upon receiving Yeastar TA100/200 gift box, please open the package and check if all the items are supplied as TA100/200 Packing List. If there is any problem, please contact your provider.

Table 2-1 TA100/200 Packing List

Item	Unit	QTY	Description
TA100/200	PC	1	TA100/200 device unit.
USB power adapter	PC	1	
USB cable	PC	1	
Network cable	PC	1	
Warranty card	PC	1	With Serial Number printed for Repair & Return.
Quick installation guide	PC	1	

Specifications and Operating Environment

Table 2-2 Specifications and Operating Environment

TA100/200	Description
Size (L×W×H)	85 mm × 58 mm × 24 mm
Power Supply	DC 5V,1A
Operating Temperature	0°C to 40°C, 32°F to 104°F
Storage Temperature	-20°C to 65°C, 4°F to 149°F
Humidity	10% to 90% (non-condensing)

Connecting Your TA100/200

Yeastar TA100/200 is designed for easy configuration and easy installation.

- **Connection of Ethernet Port**

Insert the Ethernet cable into the LAN port of TA100/200 and connect the other end of the Ethernet cable to an uplink port (a router or a switch, etc.)

- **Connection of FXS Ports**

Connect one end of a RJ11 phone cable to the FXS port, connect the other end to the analog phone.

- **Power Connection**

Connect TA100/200 to a power outlet using the included USB cable and USB power adapter.

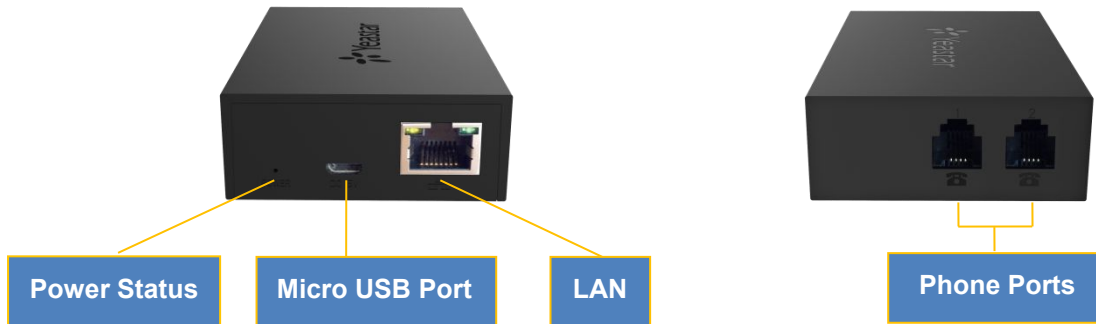


Figure 2-1 TA200 Interfaces

Table 2-3 Description of TA100/200 Connectors

Port	Description
Power Status	Power indicator.
Micro USB Port	DC 5V, 1A, for connection to power supply.
LAN Port	For connection to your router or broadband network device.
Phone Ports	For connection to analog phones/fax machines.

Getting Started

In this chapter, we guide you through the basic steps to start with a new TA100/200:

- [Accessing Web GUI](#)
- [Web Configuration Panel](#)
- [Making and Receiving Calls](#)

Accessing Web GUI

The TA100/200 attempts to contact a DHCP server in your network to obtain valid network settings (e.g., the IP address, subnet mask, default gateway address and DNS address) by default.

Please enable DHCP Server in your network to obtain the TA100/200 IP address.

How to check TA100/200 IP address:

1. Pick up the analog phone, then access the voice menu prompt by dialing “***”.
2. Dial “1” to check the IP address.
3. Dial “2” for web access address.

After entering the IP address in the web browser, users will see a log-in screen.

Check the default settings below:

Username: **admin**

Password: **password**

VoIP Analog Gateway for Cost Reduction

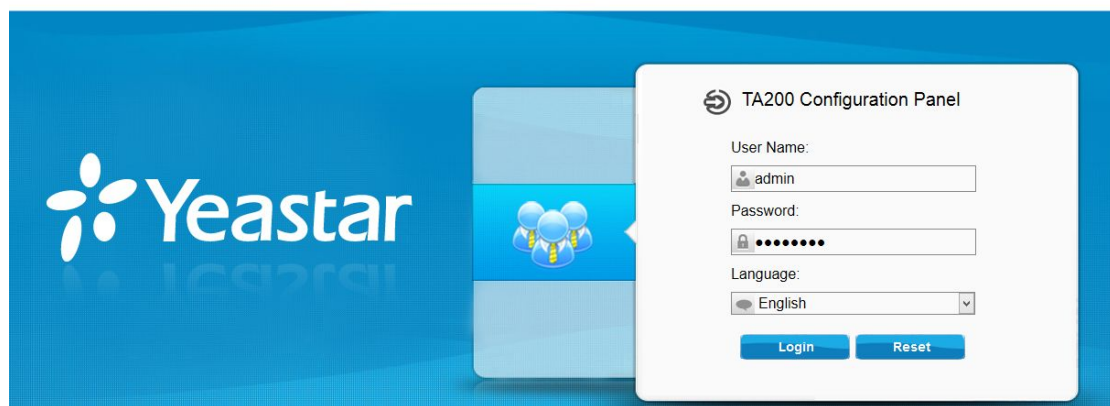


Figure 3-1 TA100/200 Login page

Web Configuration Panel

There are 4 main sections on the Web Configuration Panel for users to check the TA100/200's status and configure it.

- **Status:** check System Status, Extension Status, Trunk Status, Network Status and check call logs, system logs.
- **System:** configure Network Settings, Security related Settings, System Date and Time, Password, Backup and Restore, etc.
- **Gateway:** configure FXS ports, gateway settings and SIP settings, etc.
- **Logout:** log out TA100/200.

Note:

After saving the changes, remember to click the “Apply changes” button on the upper right corner of the Web GUI to make the changes take effect.

Making and Receiving Calls

You will need an active VoIP account from a VoIP service provider or PBX, which provides a VoIP telephone number to allow you to make and receive calls.

- **Making internal calls (for TA200 only)**
To place an internal call, pick up the analog phone and dial the other port's caller ID number.
For example, to reach the other port with caller ID number 300, dial “300”.
- **Making outbound calls**
To make an outbound call, you need to dial according to the FXS port dial pattern. By the default dial pattern, you can dial the desired outgoing number directly.
- **Answering calls**
To answer a call, pick up the handset as you usually do.

Basic Operations

In this chapter, we give instructions about how to operate on analog phones connected to TA100/200 to use some features.

- Voice Menu
- Call Hold
- Call Waiting
- Call Transfer
- Three-party Conference
- Direct IP Call
- Change TA100/200's IP Address

Voice Menu

TA100/200 provides a voice menu to guide you to configure the network settings for the device. You need to press *** on the analog phone which is connected to TA100/200's FXS port to enter the voice menu.

The default password to enter "Advanced Settings" is 123456. You can change the password on TA100/200 Web page.(Gateway→Gateway Settings→Feature Codes→Voice Menu Password Settings)

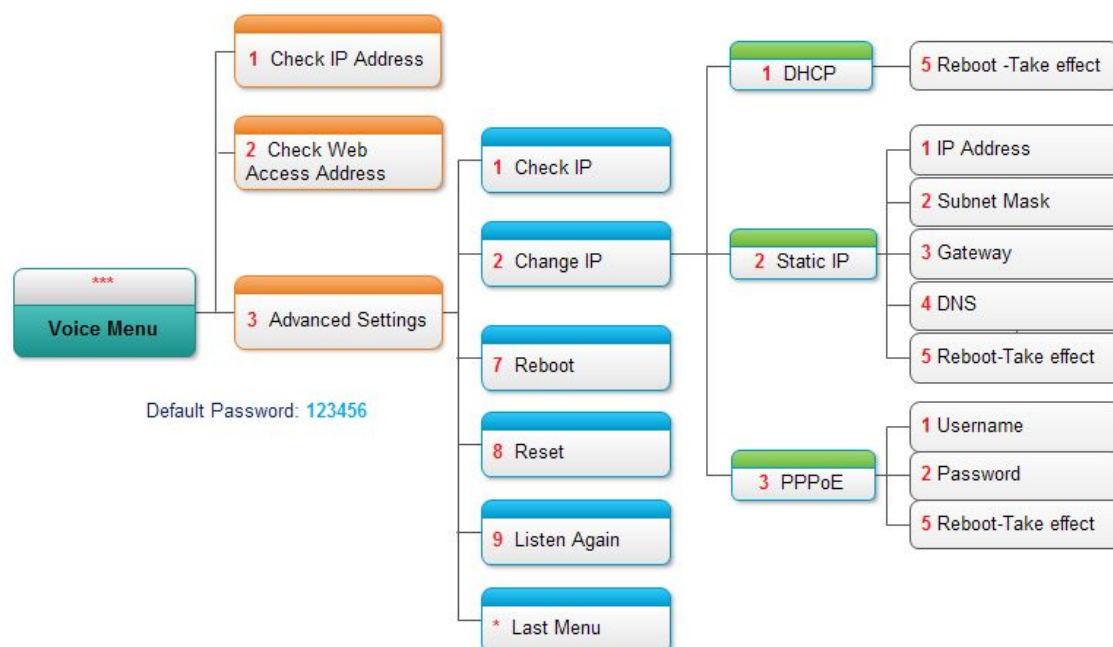


Figure 4-1 TA100/200 Voice Menu

Tips:

1. Press "9" to listen to the prompt again.
2. Press "*" to return to the last menu.

Call Hold

An active call can be held by pressing "flash" key on the analog phone. Press the key again to resume the call.

If there is no "flash" key on the phone, you can use "hook flash" (quickly toggle on-off hook) to hold a call. The call may be disconnected by chance if using "hook flash".

Call Waiting

If the call waiting is activated for the FXS port, the FXS user who is in a call can hear a call waiting tone "beep" when there is a new incoming call. The user can press "hook flash" to toggle between the active call and the incoming call.

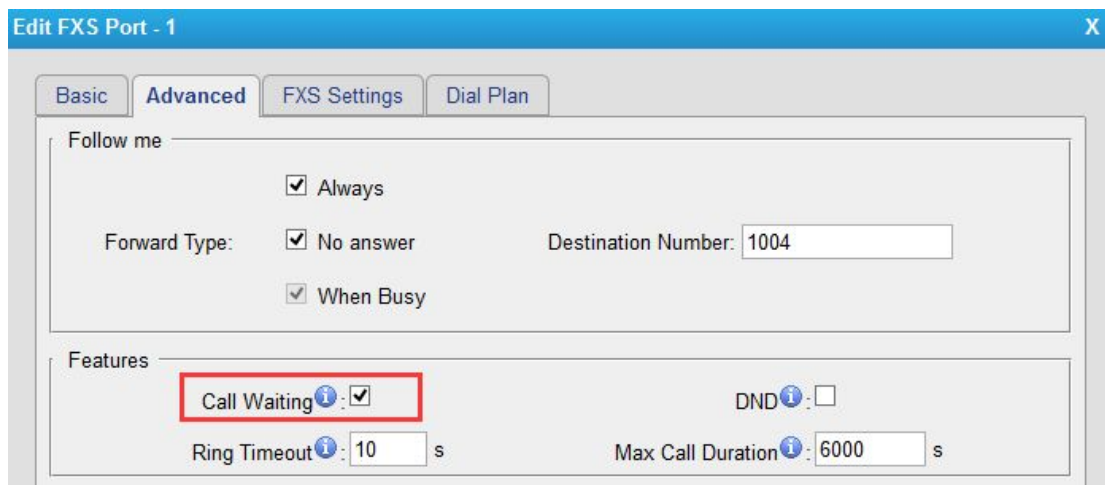


Figure 4-2 Enable Call Waiting

Call Transfer

Blind transfer and attended transfer are supported on TA100/200. Users can achieve call transfer by pressing the feature code during the call.

Blind Transfer

Default feature code: *03

1. Dial "*03" during the call;
2. Dial the called number after hearing a prompt "transfer";
3. The call will be transferred after the number is dialed.

Attended Transfer

Default feature code: *3

1. Dial "*3" during the call;
2. Dial the called number after hearing a prompt "transfer";
3. Talk to the transfer recipient;
4. The call will be transferred after hanging up.

Three-party Conference

Users can make a three-party conference call on TA100/200.

Assuming that A and B are in the call and B wants to invite C to a conference. Please check the following steps of how to establish a conference.

1. B presses "flash" key or taps hook flash to get a dial tone; A will hear the on hold music meanwhile;
2. B dials C's number;
3. If C answers the call, then B presses "flash" key or tap hook flash, the conference will be established, including A, B, and C.
4. If there is no answer on C, A can press "flash" key or tap hook flash to resume the call with A.
5. C will be ejected if B presses "flash" key or taps flash hook during the conference call.

Direct IP Call

Direct IP call allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. The default "Direct IP Calling" feature code is *96.

Example:

Target IP address: 192.168.2.123

Destination port: 5060

To call the IP phone, you should dial *96192*168*2*123*5060 on the analog phone.

Change TA100/200's IP Address Using Analog Phones

By default, TA100/200 obtains a dynamic IP address from the DHCP server. You can change the device's IP address via the analog phone which is connected to the FXS port.

There are 3 modes supported on TA100/200 access the internet.

- DHCP
- Static IP Address
- PPPoE

Here we introduce how to set a static IP address for TA100/200.

IP address: 192.168.10.125

Subnet mask: 255.255.255.0

Gateway: 192.168.10.1

DNS: 8.8.8.8

1. Press *** to enter the voice menu.
2. Press 3 to enter the "Advanced Settings".
3. Enter the password follow by the pound key: 123456# (The default password is 123456).
4. Press 2 to change the IP address.
5. Press 2 to enable the static IP.
6. Press 1 to change the IP address and follow by the new IP address (1192*168*10*125).
7. Press 2 to change the subnet mask and follow by the new subnet mask (2255*255*255*0).
8. Press 3 to change the gateway and follow by the new gateway (3192*168*10*1).
9. Press 4 to change the DNS and followed by the new DNS (48*8*8*8).
10. Press 5 to reboot the device.
11. After reboot, you can access the device by the new IP address.

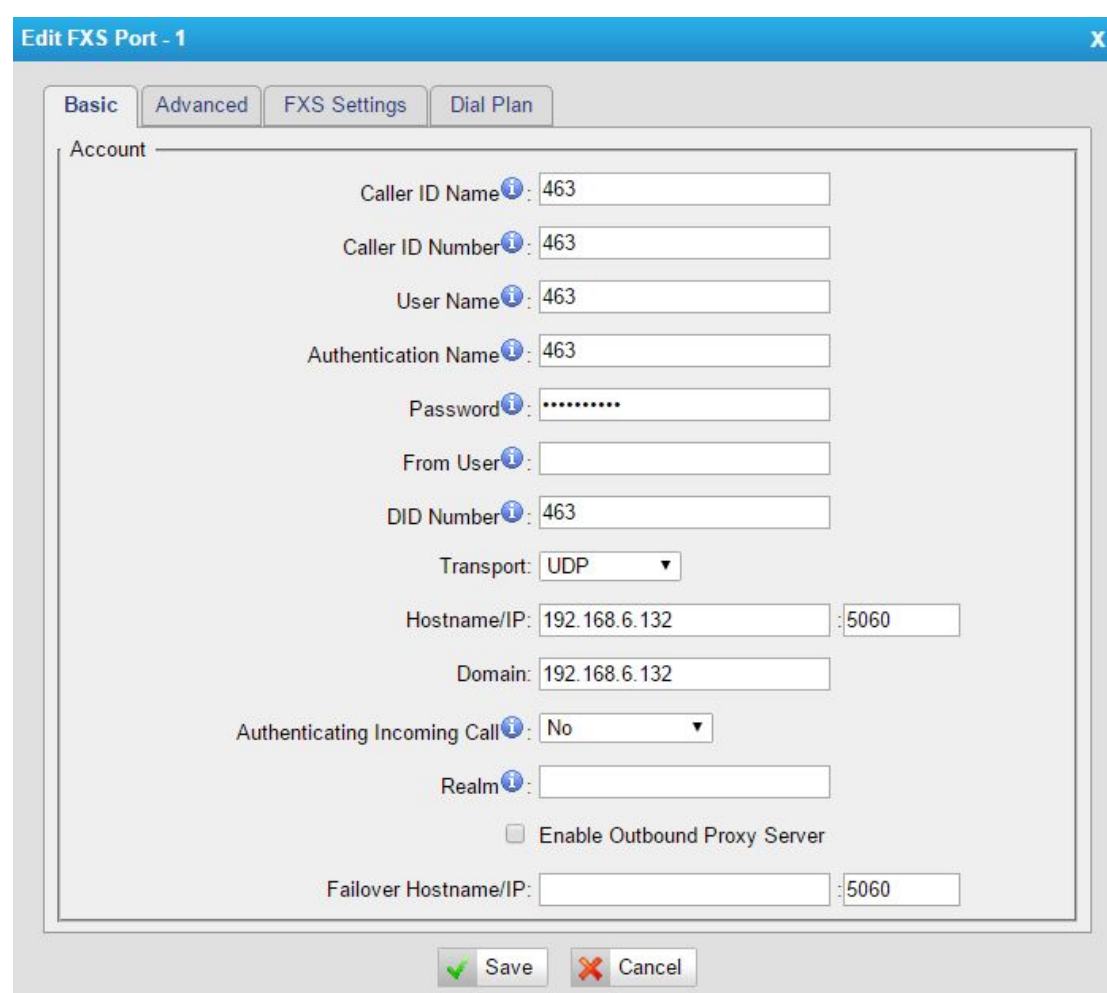
Port Settings

This chapter explains how to configure FXS port on TA100/200, go to **Gateway**→**Port Settings**→**FXS Port** page to configure the FXS ports.

- [General Settings](#)
- [Advanced Settings](#)
- [FXS Settings](#)
- [Dial Plan](#)

Click "Edit" button  to configure the FXS port.

General Settings



The screenshot shows a web-based configuration window titled "Edit FXS Port - 1". It has four tabs: "Basic", "Advanced", "FXS Settings", and "Dial Plan". The "Basic" tab is selected. Inside the "Basic" tab, there is a section labeled "Account" with the following fields:

- Caller ID Name: 463
- Caller ID Number: 463
- User Name: 463
- Authentication Name: 463
- Password:
- From User: (empty)
- DID Number: 463
- Transport: UDP (dropdown)
- Hostname/IP: 192.168.6.132 : 5060
- Domain: 192.168.6.132
- Authenticating Incoming Call: No (dropdown)
- Realm: (empty)
- ☐ Enable Outbound Proxy Server
- Fallover Hostname/IP: (empty) : 5060

At the bottom of the window, there are two buttons: "Save" (with a green checkmark icon) and "Cancel" (with a red X icon).

Figure 5-1 FXS Port General Settings

Table 5-1 Description of FXS Port General Settings

General Settings	
Caller ID Name	A character-based name for the user. For example, Bob.
Caller ID Number	The Caller ID string used when this user calls another user.
User Name	User name provided by the VoIP provider.
Authentication Name	Authentication name provided by the VoIP provider.
Password	Authentication password provided by the VoIP provider.
From User	Provided by the VoIP provider, leave this field blank if not required.
DID Number	Defines the expected DID number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info.
Transport	Choose the transport: <ul style="list-style-type: none"> • UDP • TCP • TLS
Hostname/IP	SIP Server's IP address or Domain name provided by VoIP service provider.
Domain	SIP Server's IP address or Domain name provided by VoIP service provider.
Authenticating Incoming Call	Authenticating incoming call.
Realm	A realm defines the protection space. It must be globally unique; usually it is the same with domain name. For example, when registering to China Mobile, the realm value should be "ims.fj.chinamobile.com". Fill in a realm, so that the password will be encrypted in the configuration file.
Enable Outbound Proxy	IP address or Domain name of Outbound Proxy. A proxy that receives requests from a client. Even though it may not be the server resolved by the Request-URI.
Failover Hostname/IP	Set the failover server for the account. This server will be used if the primary server is unavailable.

Advanced Settings

Edit FXS Port - 1

Basic

Advanced

FXS Settings

Dial Plan

Follow me

☐ Always

Forward Type:

☒ No answer

☒ When Busy

Destination Number:

Features

Call Waiting☒

DND☐

Ring Timeout10 s

Max Call Duration6000 s

SIP

Qualify:☐

DTMF Mode:rfc2833

Enable SRTP☐

Codec

First Codec:u-law

Second Codec:a-law

Third Codec:None

Fourth Codec:None

Fifth Codec:None

Save

Cancel

Figure 5-2 FXS Port Advanced Settings

Table 5-2 Description of FXS Port Advanced Settings

Follow Me	
Choose the forward type and configure the relevant destination number. In different conditions, the incoming calls to the account will be forwarded to different destinations.	
<ul style="list-style-type: none">Always: always forward calls to the destination number.No Answer: forward calls when no one answers the call.When Busy: forward the call when the account is busy.	
Features	
Call Waiting	Check this option if the extension should have Call Waiting capability. If this option is checked, the “When busy” follow me options will not be available.
DND	Don’t Disturb. When DND is enabled for an extension, the extension will not be available.
Ring Timeout	Check this option if you want to customize the ring time. Ring tone will stop over the time defined.
Max Call Duration	Set the max call duration for the account.

SIP	
Qualify	Whether to send check alive packets to the VoIP service provider.
DTMF Mode	Set the DTMF mode: <ul style="list-style-type: none">• rfc2833• info• inband• auto
Enable SRTP	Enable or disable SRTP.
Codec	
Set the codec for this account and its priority.	

FXS Settings

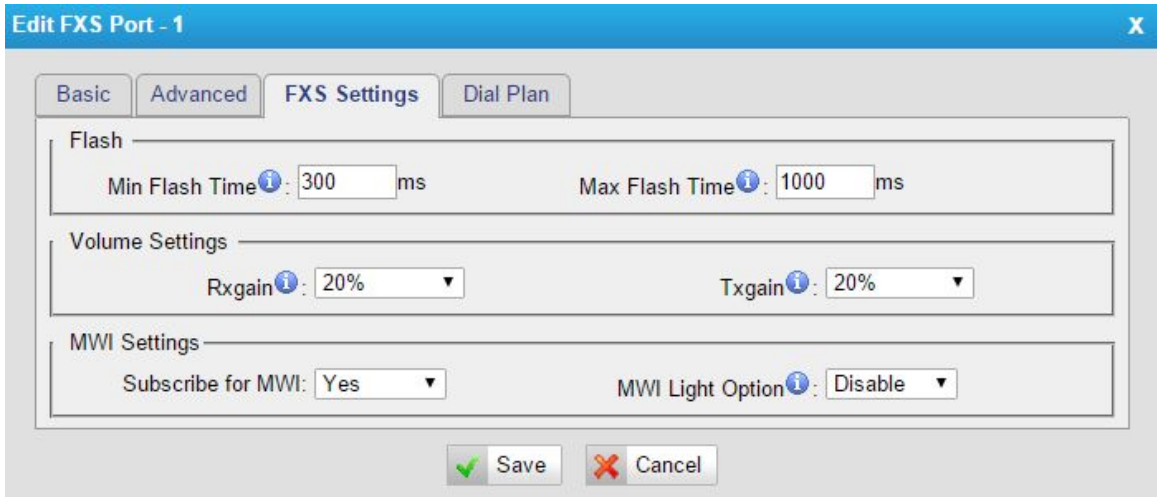


Figure 5-3 FXS Settings

Table 5-3 Description of FXS Settings

Flash	
TA100/200 will take the flash duration between Min Flash and Max Flash as an effective flash. Any flash lasting over the longest time will be considered by TA100/200 as hang-up. <ul style="list-style-type: none">• Min Flash Time: set the minimum flash time, the default is 300ms.• Max Flash Time: set the maximum flash time, the default is 1000ms.	
Volume Settings	
Rxgain	Adjust receive gain.
Txgain	Adjust transmit gain.
MWI Settings	
Subscribe for MWI	When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
MWI Light Option	To light up the MWI via FXS Reverse Polarity.

Dial Plan

Edit FXS Port - 1

Basic

Advanced

FXS Settings

Dial Plan

Hotline

Enable Hotline: No

Hotline Number:

Delay Dial: 2 s

Dial Pattern

Dial Pattern

Strip

Prepend

+ Add

Save

Cancel

Figure 5-4 Dial Plan

Figure 5-4 Description of Dial Plan Settings

Hotline	
Enable Hotline	Select whether to use Hotline or not. Hotline is disabled by default. If this feature is enabled, the system will dial out the hotline number automatically after the time of Delay Dial.
Hotline Number	Set the number to dial out automatically after the time of Delay Dial.
Delay Dial	Define how long to make Hotline take effect after you pick up the phone.
Dial Pattern	
A dial pattern defines the account can dial which numbers out. A dot is filled in this field by default.	
X	Refers to any digit between 0 and 9
Z	Refers to any digit between 1 and 9
N	Refers to any digit between 2 and 9
[###]	Refers to any digit in the brackets, example [123] is 1 or 2 or 3. Note that multiple numbers can be also separated by dots and ranges of numbers can be specified with a dash ([1.3.6-8]) would match the numbers 1, 3, 6, 7 and 8.
. (dot)	Wildcard. Match any number of anything.
!	Used to initiate call processing as soon as it can be determined that no other matches are possible.
Strip	
Allow the users to specify the number of digits that will be stripped from the front of the phone number before the call is placed.	

For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.

Prepend

Digits to prepend to a successful match. If the dialed number matches the patterns, then this will be prepended before sending to the trunks.

For example if a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before the calls are placed. When using analog trunks, a “w” character may also be prepended to provide a slight delay before dialing.

Gateway Settings

This chapter explains Gateway settings, which can be applied globally to TA100/200. The gateway settings can be configured under **Gateway**→ **Gateway Settings**.

- Basic Preferences
- Feature Codes
- Speed Dial
- System Prompts

Basic Preferences

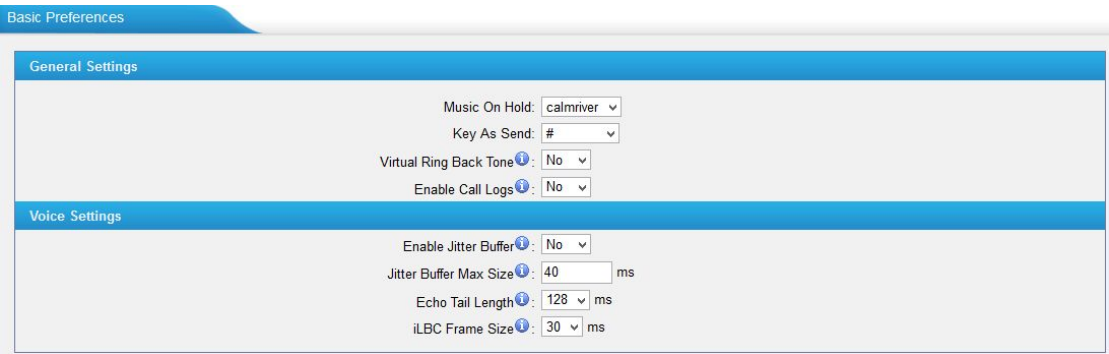


Figure 6-1 Basic Preferebces

Table 6-1 Description of Basic Preferences

General Settings	
Key As Send	Set the "#" or "*" to perform as a send key while dialing. Default is "#". Or you can disable it.
Virtual Ring Back Tone	Once enabled, when the caller dials out, the caller will only hear the virtual ring back tone generated by the system before the callee answers the call.
Enable Call Logs	Whether to store call logs in the system: <ul style="list-style-type: none">• Yes: the system will store the call logs.• No: the call logs will not be kept anymore, but the call logs stored previously will still be there.
Voice Settings	
Enable Jitter Buffer	Forces the use of a jitter buffer on the received side of a SIP channel. The call quality and fluency will be improved if this option is enabled.
Jitter Buffer Max Size	Max length of the jitter buffer. Default is 40 milliseconds. The more size, the more delay of the voice have, but will have better fluency of the voice. Usually adjust to a proper number as you experienced.
Echo Tail Length	The length of time that the echo canceller stores its

	approximation of an echo in memory. It is the maximum echo delay that an echo canceller is able to eliminate.
--	---

Feature Codes

There are various feature codes on TA100/200. The feature codes are used to acquire the gateway info or activate and inactivate supplementary services. The default feature codes are illustrated below. The parameters for feature codes are configurable.

➤ General

Table 6-2 Description of General Feature Code

Items	Default	Description
Speed Dial Prefix	*98	The prefix number for applying a speed dialing. The prefix should be added ahead of the speed dial number.
Direct IP Calling	*96	Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. The default "Direct IP Calling" feature code is *96. For detailed instruction, please refer to Basic Operation.
Check Number	*97	Users can check the analog phone's number by simply dialing the "Check Number" feature code on the phone. The default "Check Number" feature code is *97.
Voice Menu	***	Users may enter the voice prompt menu by pressing *** on their phone.
Voice Menu Password Settings	123456	The password of voice menu is required before entering the advanced settings. The default password is 123456.

➤ Call Forwarding Preferences

Table 6-3 Description of Call Forwarding Preferences

Items	Default	Description
Reset to Defaults	*70	Users may reset all call forwarding defaults by calling *70 on their phone.
Enabel Forward All Calls	*71	Users may enable always forward by calling *71 on their phone.
Disable Forward All Calls	*071	Users may disable always forward by calling *071 on their phone.
Enable Forward When Busy	*72	Users may enable busy forward by dialing *72 on their phone.

Disable Forward When Busy	*072	Users may disable busy forward by calling *072 on their phone.
Enable Forward No Answer	*73	Users may enable no answer forward by calling *73 on their phone.
Disable Forward No Answer	*073	Users may disable no answer forward by calling *073 on their phone.
Forward to Number	*75	Users may activate call forwarding by dialing this feature code, followed by the extension or phone number to forward all calls to this number. Note: users may activate Forward to number by dialing *75 + phone number. E.g. by dialing *75501, all calls will be forwarded to extension 501.
Enable Do Not Disturb	*77	Activate "Do Not Disturb". Once activated, the FXS port will reject all incoming calls.
Disable Do Not Disturb	*077	Disable "Do Not Disturb" for the FXS port by pressing the feature code on the phone. It will recover normal ringing upon the arrival of incoming calls.

Speed Dial

Speed Dial feature is available on TA100/200 that allowing you to place a call by pressing a reduced number of keys. There are 16 configurable Speed Dial templates available on TA100/200 ATA.

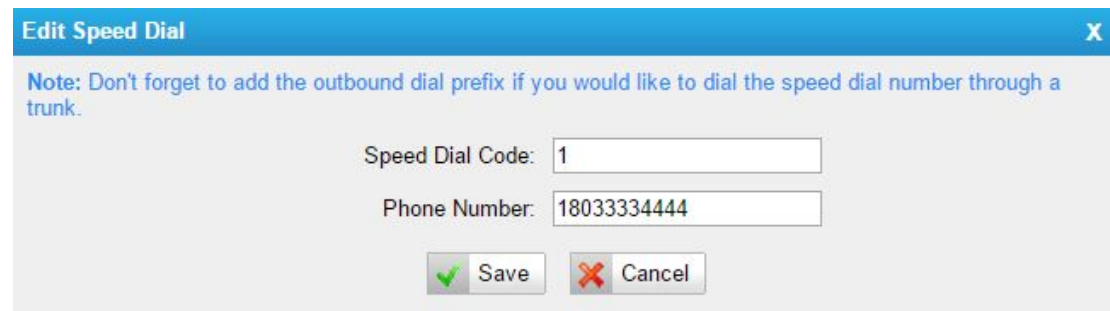


Figure 6-2 Speed Dial

- **Speed Dial Code**
The speed dialing number.
- **Phone Number**
The number you want to call.

To make a speed dial, e.g. you want to call 18033334444, simply dial *981. The *98 tells TA100/200 that you want to use the Speed Dial and the 1 is the Speed Dial Code for destination number 18033334444.

Note: don't forget to add the dial pattern according to the selected dial pattern template. That is the destination number should match the FXS port dial pattern.

System Prompts Settings

TA100/200 ships with a US English prompt set by default. The system supports English and Chinese languages. Users could update the system prompt in different ways (Auto Detection, TFTP and HTTP).

Notes:

1. Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server, please check this [address](#) for the prompts.
2. When updated successfully, just click "Apply Changes" on Web then it will take effect, there is no need to reboot.

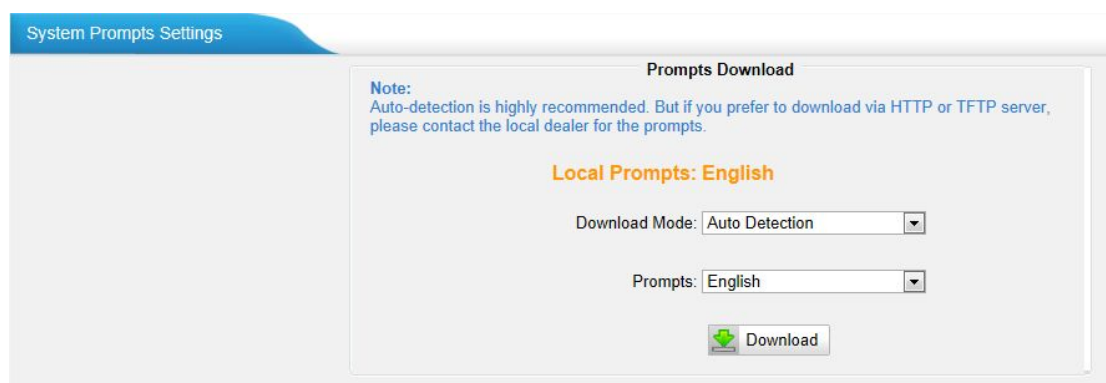


Figure 6-3 System Prompts Settings Page

Advanced Settings

This chapter explains SIP settings and Distinctive Ringtones.

- SIP Settings
- Distinctive Ringtones

SIP Settings

It is wise to leave the default setting as provided on this page. However, for a few fields, you need to change them to suit your situation.

1) General

SIP Settings

General

NAT

QOS

Response Code

Advanced Settings

UDP Port ⓘ

5060

Enable Random Port ⓘ

No ▾

☐

Enable

TCP Port ⓘ

5060

☐

Enable

TLS Port ⓘ

5061

TLS Verify Server ⓘ

No ▾

TLS Ignore Common Name ⓘ

Yes ▾

TLS Client Method ⓘ

sslv2 ▾

RTP Port Start ⓘ

10000

RTP Port End ⓘ

12000

DTMF Mode ⓘ

rfc2833 ▾

Max Registration/Subscription Time ⓘ

3600

Min Registration/Subscription Time ⓘ

60

Default Incoming/Outgoing Registration Time ⓘ

120

MWI Subscription Period ⓘ

3600

Register Attempts ⓘ

0

Register Timeout ⓘ

20

Calling Channel Codec Priority ⓘ

Yes ▾

DNS SRV Look Up ⓘ

No ▾

User Agent ⓘ

G.729 License Key ⓘ

Figure 7-1 SIP General Settings

Table 7-1 Description of SIP General Settings

Items	Description
UDP Port	Port used for SIP registrations. The default is 5060.
Enable Random Port	Enable or Disable Random SIP port.

Random Port Update Interval	Set the Random Port Update Interval.
TCP Port	Port used for SIP registrations. The default is 5060.
TLS Port	Port used for SIP registrations. The default is 5061.
TLS Verify Server	When using TA100/200 as a TLS client, whether or not to verify server's certificate. It is "No" by default.
TLS Verify Client	When using TA100/200 as a TLS server, whether or not to verify client's certificate. It is "No" by default.
TLS Ignore Common Name	Set this parameter as "No", then common name must be the same with IP or domain name.
TLS Client Method	When using TA100/200 as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.
RTP Port Start	Beginning of the RTP port range.
RTP Port End	End of the RTP port range.
DTMF Mode	Set the default mode for sending DTMF. Default setting: rfc2833
Max Registration/Subscription Time	Maximum duration (in seconds) of a SIP registration. The default is 3600 seconds.
Min Registration/Subscription Time	Minimum duration (in seconds) of a SIP registration. The default is 60 seconds.
Default Incoming/Outgoing Registration Time	Default Incoming/Outgoing Registration Time: the default duration (in seconds) of incoming/outgoing registration.
MWI Subscription Period	Duration (in seconds) of MWI subscription.
Register Attempts	The number of SIP REGISTER messages to send to a SIP Registrar before giving up. The default is 0 (no limit).
Register Timeout	Number of seconds to wait for a response from a SIP Registrar before classifying the register has timed out. The default is 20 seconds.
Calling Channel Codec Priority	Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected preferentially. If not, TA100/200 will follow the priority order in your SIP/SPS trunks.
DNS SRV Look Up	Please enable this option when your SIP trunk proxy contains more than one IP address.
User Agent	To change the user agent.
G.729 Licenses Key	If you would like to use G.729, please enter your license key.

2) NAT

SIP Settings

General

NAT

Codecs

QOS

Response Code

Advanced Settings

Note: Configuration of this section is only required when you use remote extensions.

Enable STUN:

☐

STUN Address:

STUN Port:

External IP Address:

External Host:

External Refresh Interval:

Local Network Identification:

NAT Mode:

yes

Allow RTP Re-invoke:

yes

Figure 7-2 NAT Settings

Table 7-2 Description of NAT Settings

Items	Description
Enable STUN	STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.
STUN Address	The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.
External IP Address	The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.
External Host	Alternatively you can specify an external host, and the system will perform DNS queries periodically. This setting is only required when your public IP address is not static. It is recommended that a static public IP address is used with this system. Please contact your ISP for more information.
External Refresh Interval	If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.
Local Network Identification	Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall. Some examples of this are as follows: "192.168.0.0/255.255.0.0": All RFC 1918 addresses are local networks; "10.0.0.0/255.0.0.0": Also RFC1918; "172.16.0.0/12":Another RFC1918 with CIDR notation; "169.254.0.0/255.255.0.0": Zero conf local network. Please refer to RFC1918 for more information.
NAT Mode	Global NAT configuration for the system; the options for this setting are as follows: Yes = Use NAT. Ignore address information in the SIP/SDP headers and

	reply to the sender's IP address/port. No = Use NAT mode only according to RFC3581. Never = Never attempt NAT mode or RFC3581 support. Route = Use NAT but do not include rport in headers.
Allow RTP Reinvite	By default, the system will route media steams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.

3) QoS

QoS (Quality of Service) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic. When the network capacity is insufficient, QoS could provide priority to users by setting the value.

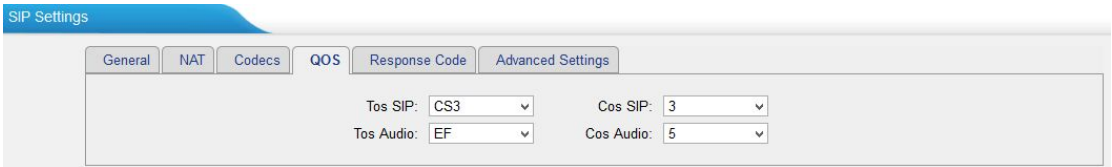


Figure 7-3 Qos

Note: it's recommended that you configure the QoS in your router or switch instead of TA100/200 side.

4) Response Code

You can change the response code on TA100/200 to the one you want before sending it to the VoIP server. It helps the VoIP server understands better the exact call status, like busy, no response and others.

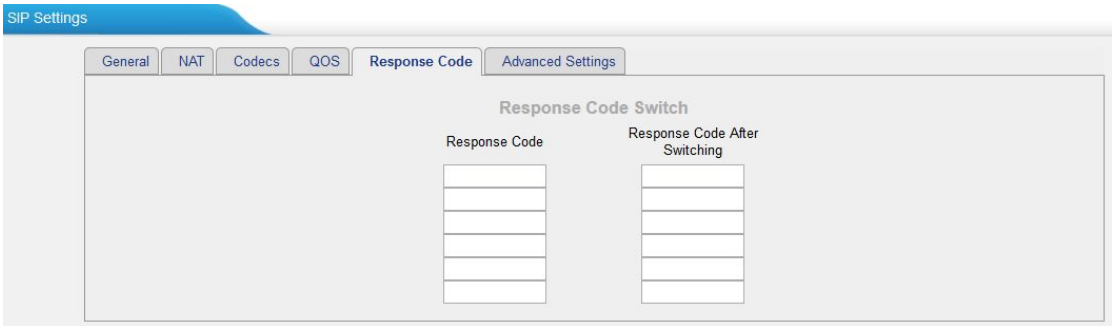


Figure 7-4 Response Code

Note: we don't recommend configuring this if you are not familiar with the code of call status from the VoIP server.

6) Advanced Settings

SIP Settings

General NAT Codecs QOS Response Code **Advanced Settings**

Call ID Field: From

DID Field: To

180 Ringing: ☐

Remote Party ID: ☐ send ☐ trust

Allow Guest: No

Pedantic: No

Alwaysauthreject: Yes

OPTIONS Response 200: Yes

Session-timers: Accept

Session-expires: 1800 s

Session-minse: 90 s

Session-refresher: Uas

Figure 7-5 SIP Advanced Settings

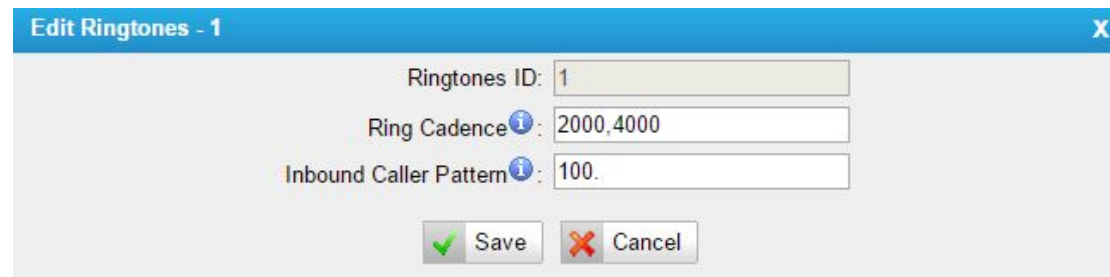
Table 7-3 Description of SIP Advanced Settings

Items	Description
Caller ID Field	Decide the caller ID will be extracted from which field of the SIP packets.
DID Field	Decide the DID number will be extracted from which field of the SIP packets.
180 Ringing	It is set when the telecom provider needs. Usually it is not needed.
Remote Party ID	Whether to send Remote-Party-ID on SIP header or not. Default: no.
Allow Guest	Whether to allow anonymous registration extension or not. Default: no. It's recommended that it is disabled for security reason.
Pedantic	Enable pedantic parameter. Default: no.
Alwaysauthreject	If enabled, when TA100/200 rejects incoming "Register" or "Invite" packets, TA100/200 always respond the packets using "SIP 403 Forbidden". It's recommended that it is enabled for security reason.
OPTIONS Response 200	Whether to respond "200 OK" to an OPTION message.
Session-timers	Enable session-timer mode, default: yes. If you find the call is cut off every 15 minutes every time, please disable this.
Session-expires	The max refresh interval
Session-minse	The min refresh interval, which mustn't be shorter than 90s.
Session-refresher	Choose the session-refresher, the default is Uas.

Distinctive Ringtones

TA100/200 provides 10 configurable distinctive ringtones. Users could configure different ringtones to match different incoming caller ID.

For example, if Inbound Caller Pattern is configured as “100.”, all the incoming calls start with digits “100” will ring using cadence “2000,4000” as the following figure shows.



The screenshot shows a dialog box titled "Edit Ringtones - 1" with a close button (X) in the top right corner. Inside the dialog, there are three labeled input fields: "Ringtones ID" containing the number "1", "Ring Cadence" containing the text "2000,4000", and "Inbound Caller Pattern" containing the text "100.". Each label has a small blue information icon to its right. At the bottom of the dialog, there are two buttons: "Save" with a green checkmark icon and "Cancel" with a red X icon.

Figure 7-6 Distinctive Ringtones

Network Preferences



This chapter explains network settings on TA100/200. Click the main menu **System** on the top of the Web GUI to check the network settings.

- LAN Settings
- Service
- VLAN Settings
- VPN Settings

LAN Settings

After successfully logging in the TA100/200 Web GUI for the first time, users could go **System→Network Preferences→LAN Settings** to configure the network for TA100/200.

Figure 8-1 LAN Settings

Table 8-1 LAN Settings

Items	Description
Hostname	Set the host name for TA100/200.
Mode	Choose the network mode: <ul style="list-style-type: none">• Static IP Address• DHCP• PPPoE
IP Address	Set the IP Address for TA100/200.
Subnet Mask	Set the subnet mask for TA100/200.
Gateway	Set the gateway for TA100/200.
Primary DNS	Set the primary DNS for TA100/200.
Secondary DNS	Set the secondary DNS for TA100/200.
IP Address2	Set the second IP Address for TA100/200.
Subnet Mask2	Set the second subnet mask for TA100/200.

LAN Settings

General Settings

Hostname: TA200

Mode: DHCP

Figure 8-2 DHCP Mode

Select DHCP mode to get network automatically from the local network.

LAN Settings

General Settings

Hostname: TA200

Mode: PPPoE

User Name:

Password:

Figure 8-3 PPPoE

Fill in user name and password to access the Internet via PPPoE.

Service

The administrator can manage all the access methods on TA on the "Service" page.

Service

General Service Settings

Enable SSH: Yes Port: 8022

Enable FTP: Yes Port: 21

Web Server

HTTP: Enabled

HTTP Bind Port: 80

HTTPS: Disabled

HTTPS Bind Port: 443

Figure 8-4 Service Settings

Table 8-2 Description of Service Settings

Items	Description
SSH	By using SSH, you can log in to TA100/200 and run commands. It's disabled by default. We don't recommend enabling it if not needed. The default port for SSH is 8022.
FTP	FTP access; The default port is 21.
HTTP	HTTP web access; The default port is 80.
HTTPS	HTTPS web access, it is disabled by default, and you can enable it to get safer web access.

VLAN Settings

VLAN (Virtual Local Area Network) is a group of hosts with a common set of requirements, which communicate as if they were attached to the same broadcast domain, regardless of their physical location.

A VLAN is a broadcast domain created by switches. This means the VLAN is configured on switches, layer 3 switches. Note that some of the switches don't support VLAN.

Note:

TA100/200 acts as a VLAN client, a 3-layer switch is needed.

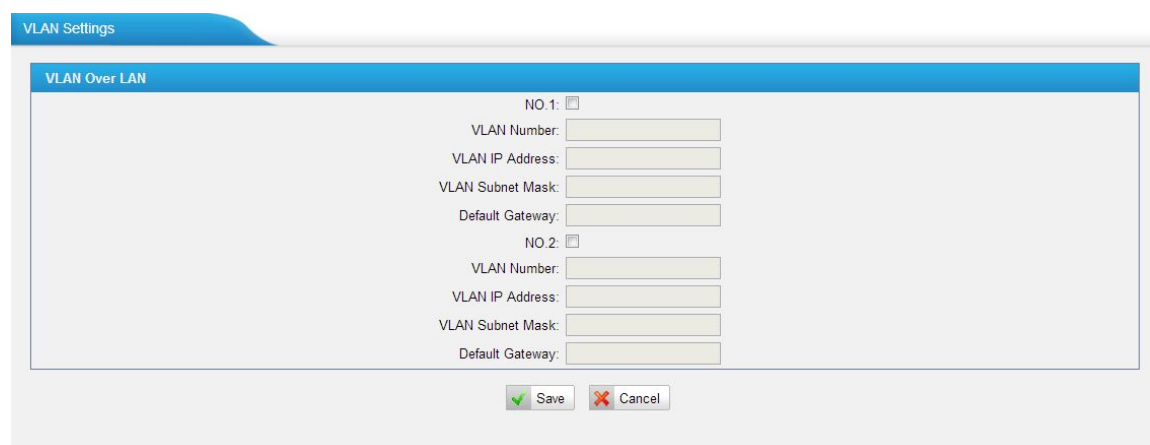


Figure 8-5 VLAN Settings

Please follow the steps below to set up VLAN on TA100/200.

Step1. Create VLANs on your switch.

Step2. Allocate a VLAN ID and IP address for TA100/200.

Step3. Configure VLAN settings page on TA100/200.

VPN Settings

A virtual private network (VPN) is a method of computer networking typically using the public internet that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. TA100/200 supports OpenVPN.

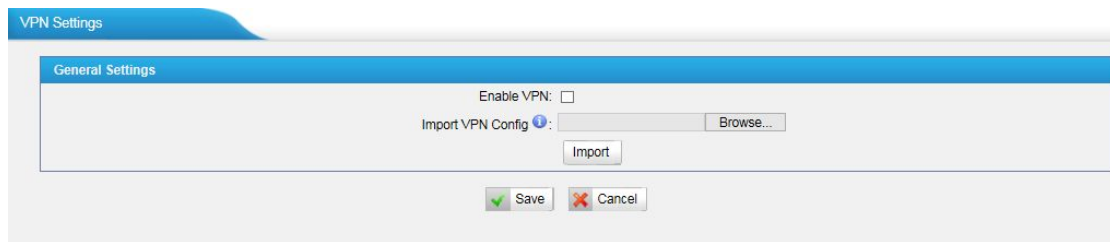


Figure 8-6 VPN Settings

- **Enable VPN**
Enable VPN feature.
- **Import VPN Config**
Import configuration file of OpenVPN.

Notes:

1. Uncomment “user” and “group” in the “config” file. You can get the config package from the OpenVPN provider.
2. TA100/200 works as VPN client mode only.

Security Center

This chapter describes how to secure your TA100/200. It is strongly recommended that users configure firewall and other security options on TA100/200 to prevent the attack fraud and the system failure or calls loss.

- Security Center
- AMI Settings
- Certificates
- Firewall Rules
- IP Blacklist

Security Center

All the security settings including Firewall, Service, Port Settings in TA100/200 are displayed in Security Center. Users could rapidly check and configure the relevant security settings here.

1) Firewall

In the “Firewall” tab, users could check firewall configuration and alert settings. By clicking the relevant button, you can enter the configuration page directly.

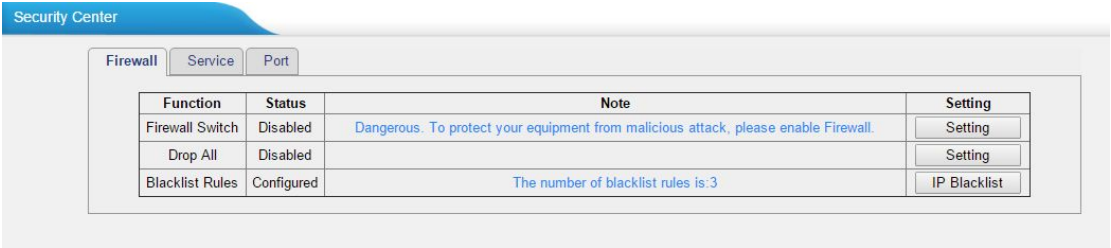


Figure 9-1 Security Center—Firewall

2) Service

In “Service” tab, you can check AMI/SSH status. For AMI/SSH, you can enter the according page by clicking the button in “Setting” column.

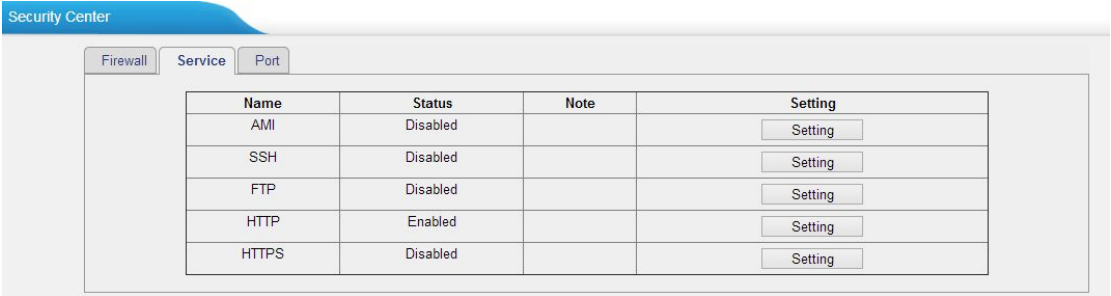


Figure 9-2 Security Center—Service

3) Port

In “Port” tab, you can check SIP port and HTTP port. You can also enter the relevant page by clicking the button in “Setting” column.

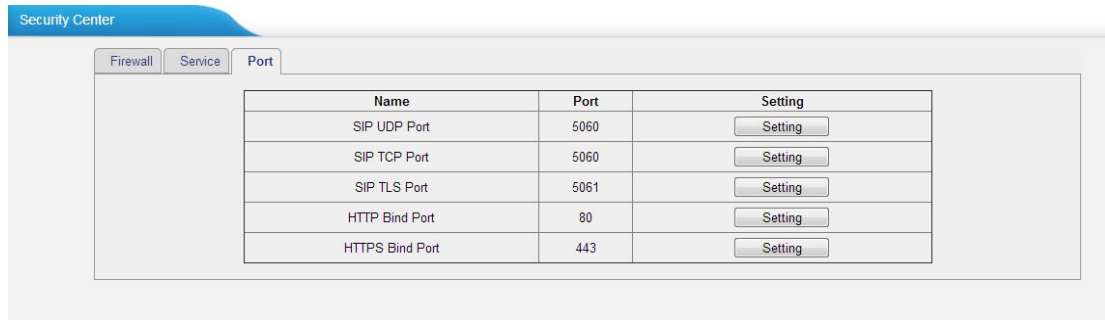


Figure 9-3 Security Center—Port

AMI Settings

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. It allows live monitoring of events that occur in the system, as well enabling you to request that Asterisk perform some action. The actions that are available are wide-ranging and include things such as returning status information and originating new calls. Many interesting applications have been developed on top of Asterisk that take advantage of the AMI as their primary interface to Asterisk.

There are two main types of messages on the Asterisk Manager Interface: manager events and manager actions.

The 3rd party software can work with TA100/200 using AMI interface. It is disabled by default. If necessary, you can enable it.

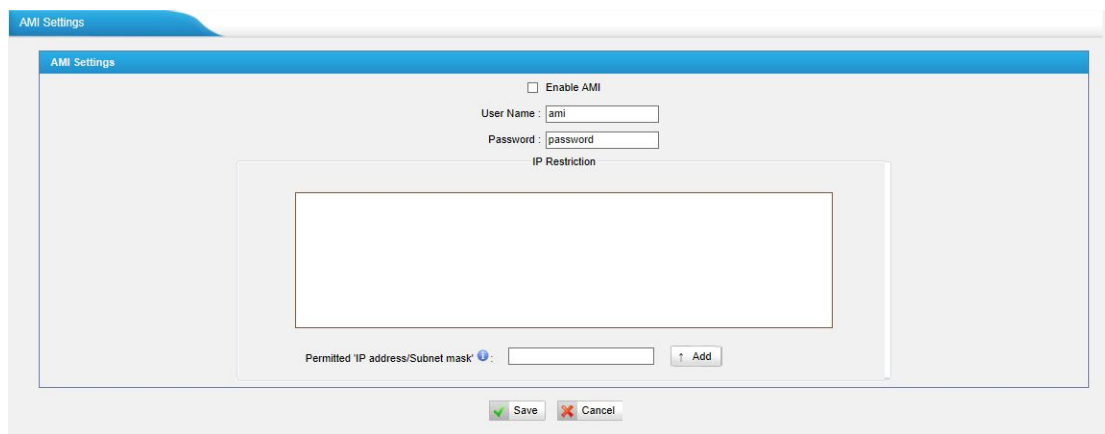


Figure 9-4 AMI Settings

- User Name, Password&Port**
 After enabling AMI, you can use this username and password to log in TA100/200. The default port is 5038.
- Permitted "IP address/Subnet mask"**

You can set which IP is allowed to log in TA100/200 AMI interface.

Certificates

TA100/200 supports TLS transport, you can configure FXS port with TLS transport. To use TLS, you should upload certificates first.

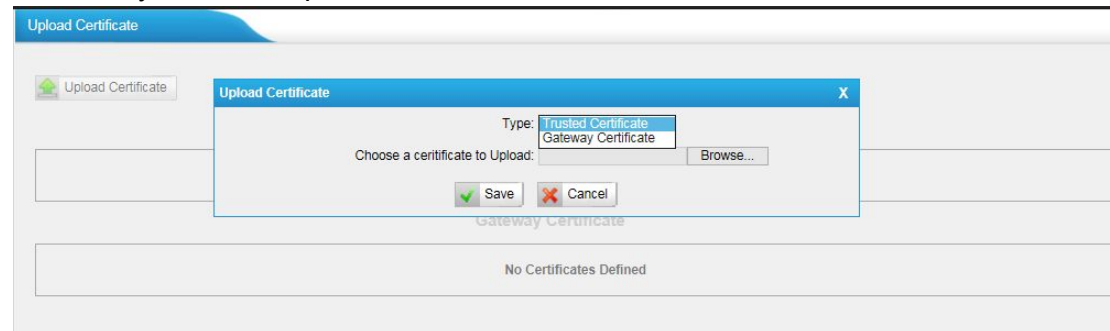


Figure 9-5 Upload Certificate

- **Trusted Certificate**
This certificate is a CA certificate. When selecting “TLS Verify Client” as “Yes”, you should upload a CA. The relevant VoIP provider should also have this certificate.
- **Gateway Certificate**
This certificate is server certificate. No matter selecting “TLS Verify Client” as “Yes” or “NO”, you should upload this certificate to TA100/200. If the VoIP provider enables “TLS Verify server”, you should also upload the relevant CA certificate on the VoIP provider.

Firewall Rules

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.



Figure 9-6 Firewall Settings

1) General Settings

Table 9-1 Description of Firewall General Settings

Items	Description
Enable Firewall	Enable the firewall to protect the device.
Disable Ping	Enable this item to drop net ping from remote hosts.
Drop All	When you enable “Drop All” feature, the system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one “TCP” accept common rule must be created for port used for SSH access, port used for HTTP access and port sued for CGI access.

2) Common Rules

There is no default rule; you can create one as required.

Add Firewall Rule

Name

Description

Protocol

UDP

Port

IP

MAC Address

Action

Drop

Figure 9-7 Common Rules

Table 9-2 Description of Common Rules

Items	Description
Name	A name for this rule, e.g. "HTTP".
Description	Simple description for this rule. E.g. accept the specific host to access the Web interface for configuration.
Protocol	The protocols for this rule.
Port	Initial port should be on the left and end port should be on the right. The end port must be equal to or greater than start port.
IP	The IP address for this rule. The format of IP address is: IP/mask E.g. 192.168.5.100/255.255.255.255 for IP 192.168.5.100 E.g. 192.168.5.0/255.255.255.0 for IP from 192.168.5.0 to 192.168.5.255.
MAC Address	The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.
Action	Accept: Accept the access from remote hosts. Drop: Drop the access from remote hosts. Ignore: Ignore the access.

Note: the MAC address will be changed when it's a remote device, so it will not be working to filter using MAC for remote devices.

3) Auto Defense

Add Auto Defense Rule

Port

Protocol

UDP

Rate

Second

Save

Cancel

Figure 9-8 Auto Defense

Table 9-3 Description of Auto Defense

Items	Description
Port	The port you want to auto defense, for example, 8022.
Protocol	Select the protocol. You can select UDP or TCP.
Rate	The maximum packets or connections can be handled per unit time. For example, if you configure it as below: Port: 8022 Protocol: TCP Rate: 10/min Then, it means maximum 10 TCP connections can be handled in 1 minute. The 11 th connection will be dropped.

IP Blacklist

You can set some packets accept speed rules here. When an IP address, which hasn't been accepted in common rules, sends packets faster than the allowed speed, it will be set as a black IP address and be blocked automatically.

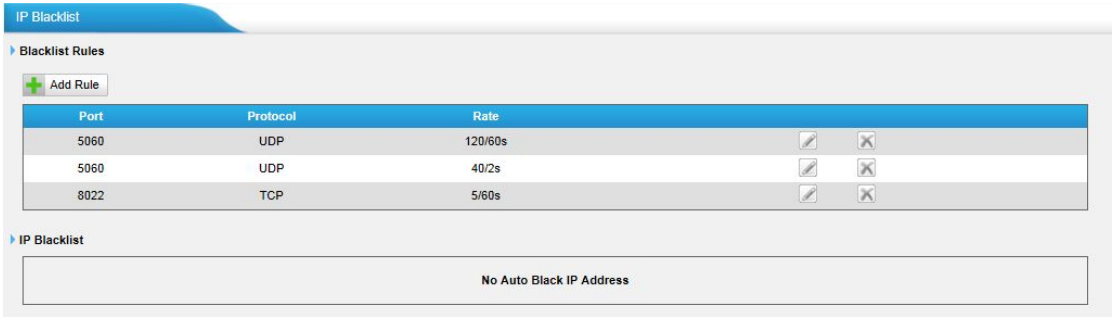


Figure 9-9 IP Blacklist Settings Page

1) Blacklist rules

We can add the rules for IP blacklist rate as demanded.

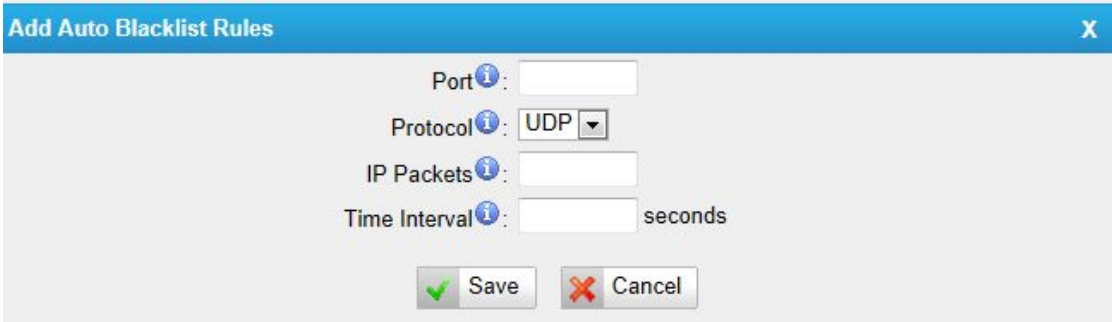


Figure 9-10 Add Blacklist Rule

Table 9-4 Description of Auto Blacklist Rules

Items	Description
Port	Auto defense port
Protocol	Auto defense protocol. TCP or UDP.
IP Packets	Allowed IP packets number in the specific time interval.
Time interval	The time interval to receive IP packets. For example, IP packets 90, time interval 60 means 90 IP packets are allowed in 60 seconds.

2) IP blacklist

The blocked IP address will display here, you can edit or delete it as you wish.

System Preferences

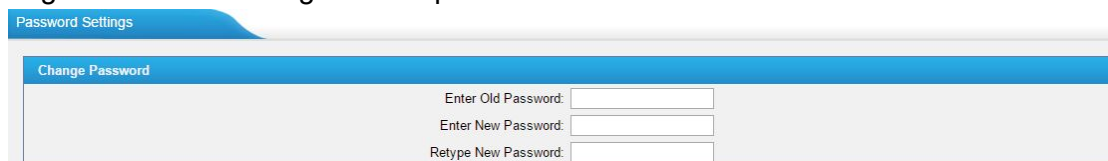
This chapter describes system maintenance settings including the followings:

- Password Settings
- Date and Time
- Auto Provision Settings
- Firmware Upgrade
- Backup and Restore
- Reset and Reboot

Password Settings

It is highly recommended to change the system's password after first login. Go to **System→System Preferences→Password Settings** to change the password.

1. Enter the old password first.
2. Enter a new password and retype the new password to confirm. The password complexity will be detected, which will help users to set a strong password and make TA100/200 safer. A strong password is comprised of letters, numbers and characters.
3. Save the changes, the user will be automatically logged out.
4. Log in TA100/200 using the new password.



The screenshot shows a web interface for 'Password Settings'. It features a blue header bar with the text 'Change Password'. Below this, there are three input fields with labels: 'Enter Old Password:', 'Enter New Password:', and 'Retype New Password:'. Each label is followed by a text input box.

Figure 10-1 Password Settings

Date and Time

Please adjust the time of TA100/200 (including the time zone) consistent with your local time. Go to **System→System Preferences→Date and Time** to configure the system date and time.

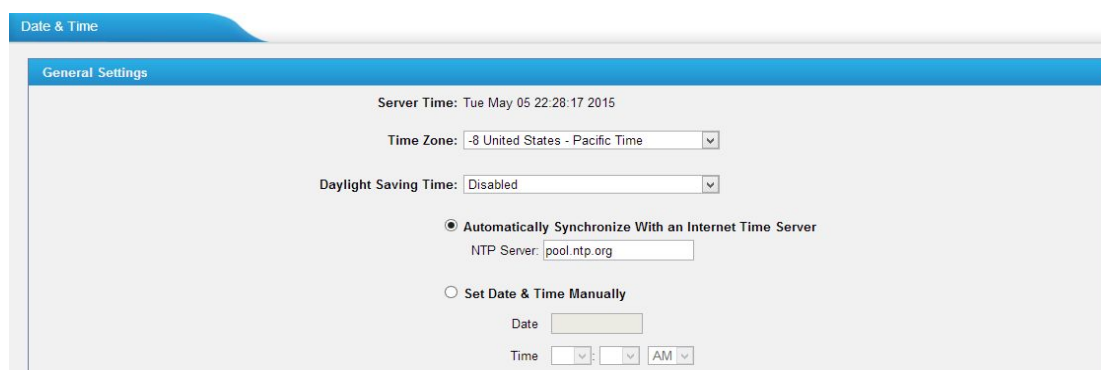


Figure 10-2 Date and Time

- **Time Zone**
Select your current and correct time zone on TA100/200.
- **Daylight Saving Time**
The option is disabled by default. Enable it when necessary.
- **Automatically Synchronize with an Internet Time Server**
TA100/200 will adjust its internal clock to a central network server. Please note the TA100/200 should be able to access to the Internet if you choose this method.
- **Set Date & Time Manually**
Enter the time using the numbers on your keyboard.

Note: you have to reboot the system to make the changes take effect.

Auto Provision Settings

Three methods are supported for Auto Provision: PNP, DHCP and you can manually configure a server URL to get the configuration file from the server. Go to **System→System Preferences→Auto Provision Settings** to configure.

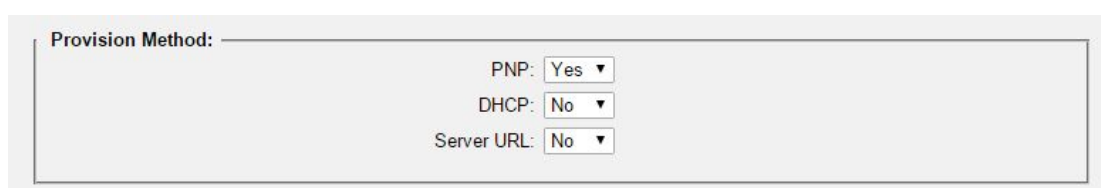


Figure 10-3 Auto Provision Methods

PNP and **DHCP** modes work along with MyPBX "TA Provisioning". Firstly, users need to configure TA100/200 on MyPBX "TA Provisioning" page. Then TA100/200 will find and get the configuration file from MyPBX during boots up.

In **PNP** mode, you just need to place the TA100/200 in the same IP range network with MyPBX, then you can find the TA100/200 and provision it on MyPBX "TA Provisioning" page.

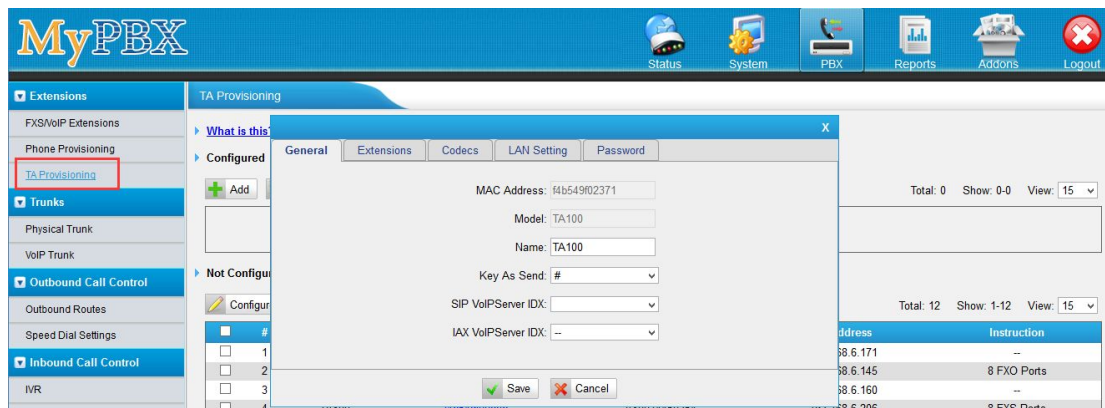


Figure 10-4 MyPBX TA Provisioning

If you use **DHCP** mode to do auto provision, you should enable DHCP Server on MyPBX to make it as a DHCP server. (System→Network Preferences→DHCP Server).

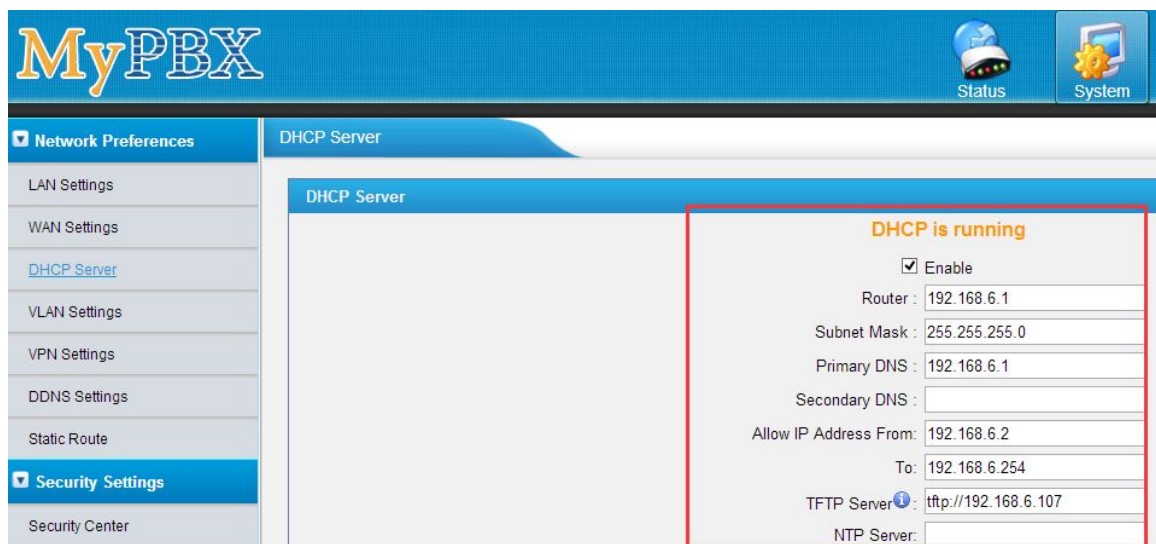


Figure 10-5 Set MyPBX as a DHCP Server

Then select DHCP mode on LAN settings page to make TA100/200 as a DHCP client.

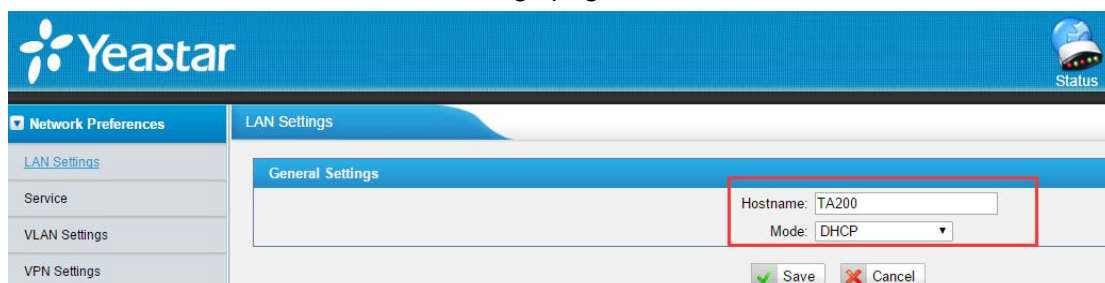


Figure 10-6 Set TA100/200 as a DHCP Client

Another way to do auto provision is to download configuration file from the configured server URL. Fill in the URL, user name, password, and set the time, TA100/200 will get the configuration file from the server automatically and regularly.

Note: if there is no user name and password for the server, leave these fields blank.

Server Settings:

Server URL :

User Name :

Password :

☐ Interval of time Minute

☒ Specified time :

Other:

AES Key :

Always Apply :

Figure 10-7 Server Address

- **AES Key**
If the configuration file is encrypted by AES key, you need to fill the key in this field.
- **Always Apply**
With No, it will compare the current configuration file with the last updated one, if the contents are the same no update will be applied. With Yes, it will always apply the updated configuration file.

Firmware Upgrade

TA100/200 can be upgraded to a new firmware version via network or locally. Users could upgrade firmware via HTTP or TFTP. Please go to **System**→ **System Preferences**→ **Firmware Update** to do upgrade.

Notes:

1. If “Reset configuration to Factory Defaults” is enabled, the system will be restored to factory default settings.
2. When updating the firmware, please don’t turn off the power. Or the system will be damaged.
3. If you are trying to upgrade through HTTP, please make sure that your TA100/200 is able to visit external network, or it cannot access Yeastar website to get the firmware file, causing the upgrade fail.

Upgrade through HTTP

On the Firmware Upgrade page, choose **HTTP URL**.

Step1. Enter the download link of the firmware file.

Note: the HTTP URL should be a **BIN** file download link.

Step2. Click “Start” to upgrade.

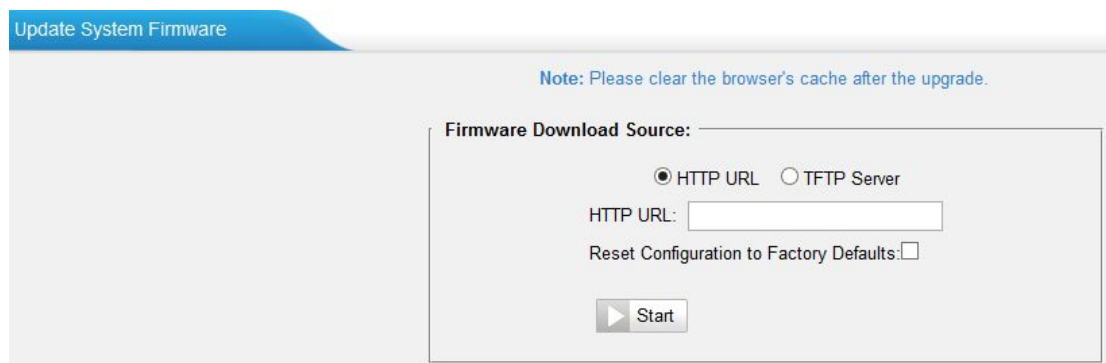


Figure 10-8 Upgrade through HTTP

Upgrade through TFTP

Step1. Download firmware file from Yeastar website.

Step2. Create a tftp Server (For example, tftpd on Windows).

- 1) Install tftpd32 software on computer.

Download link: http://tftpd32.jounin.net/tftpd32_download.html

- 2) Configure tftpd32.

On option “**Current Directory**”, click “**Browse**” button, choose the firmware file (BIN file) upgraded patch.

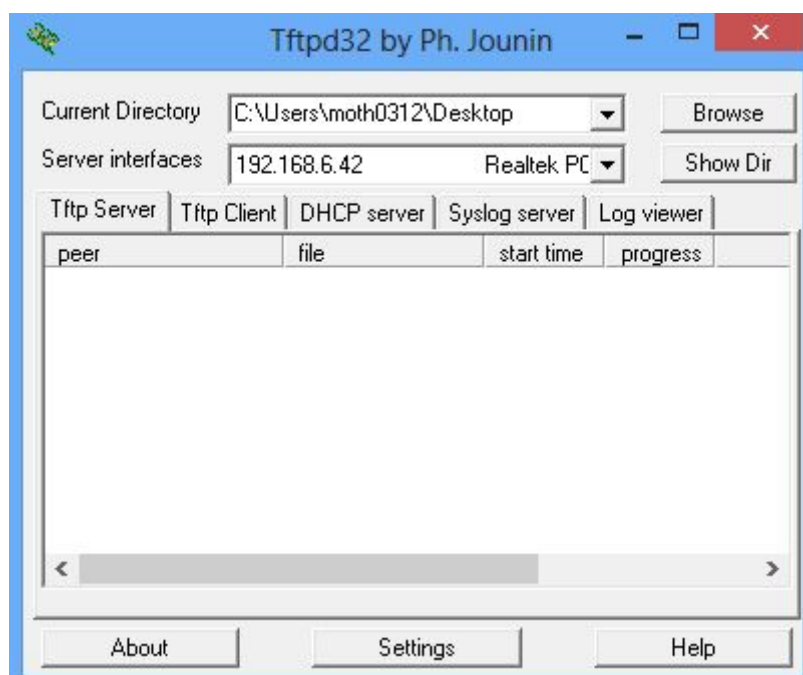


Figure 10-9 Configure Tftpd32

Step3. Logon the TA100/200's Web page and go to **System→System Preferences→Firmware Update**, choose “**TFTP Server**”.

- 1) TFTP Server: fill in IP address of tftpd32 server (your PC's IP address).
- 2) File Name: enter the name of firmware update. It should be a BIN file name.

3) Click “Start” to upgrade.



Figure 10-10 Upgrade through HTTP


Backup and Restore

TA100/200 provides Backup and Restore feature, which allows you to create a complete backup of TA100/200 configurations to a file.


Notes:

- 1. When you have updated the firmware version, it's not recommended to restore using an old package.
- 2. Backup from an earlier version cannot be restored on TA100/200 of a later version.


- **Create a New Backup**

Click  **Create a New Backup** to create a new backup.

- **Upload a Backup**

Click  **Upload a Backup** to upload a backup.

- **Restore**

To restore TA100/200 configuration data, upload the backup file to TA100/200 and click . Reboot the system to take effect.

Please note the current configurations will be **OVERWRITTEN** with the backup data.



#	Name	Time	Options
1	backup_2015may9_174120.tar	Sat May 09 1:41:58 2015	  

Figure 10-11 Restore Backup

Reset and Reboot

Users could reset and reboot the system under **System→ System Preferences→**

Reset and Reboot.

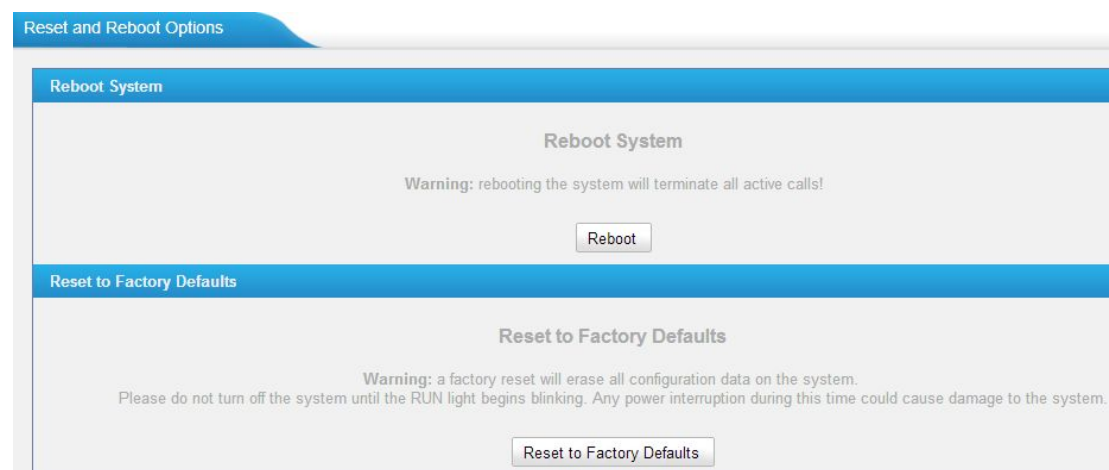


Figure 10-12 Reset and Reboot

Status

Users could check the system status on **Status→System Status**, where FXS Port Status, Network Status and System Info can be checked. CDR and System Logs can be checked under **Status→ Reports**.

- FXS Port Status
- Network Status
- System Info
- Call Logs
- System Logs
- Packet Tool

FXS Port Status

FXS Port Status						
Port	UP/Down/Break	Name	Status	Voice Mail(New/Old)	Off-hook/On-hook	Phone Status
1	Up	463	OK	--	On Hook	Connected
2	Up	460	OK	--	On Hook	Disconnected

Figure 11-1 FXS Port Status

Table 11-1 Description of FXS Port Status

Up/Down	
Up	The FXS module works well.
Down	The FXS module is broken.
Status	
OK	Successful registration, the account is ready to use.
Unregistered	Registration failed.
Request Sent	Registering.
Unreachable	The SIP server is unreachable.
Waiting for authentication	Wrong password or user name.
Voice Mail (New/Old)	
Showing the number of unread voicemail and old voicemail.	
Hook	
On Hook	The FXS port is idle.
Off Hook	The FXS port is busy.
Phone Status	
Connected	An analog phone or fax machine is connected to the port.
Disconnected	No device is connected to the port.

Network status

In this page, the IP address of LAN port will appear with their status.

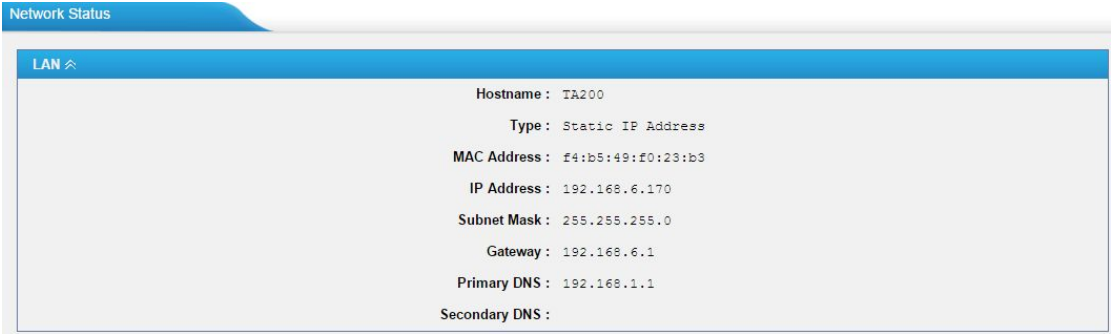


Figure 11-2 Network Status

If your VLAN or VPN are configured, you can check the status in this page also.

System Info

In this page, we can check the hardware/firmware version, or the disk usage of TA100/200.



Figure 11-3 System Info

Call Logs

The call log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by call date, caller/callee, trunk, duration, billing duration, status, or communication type.

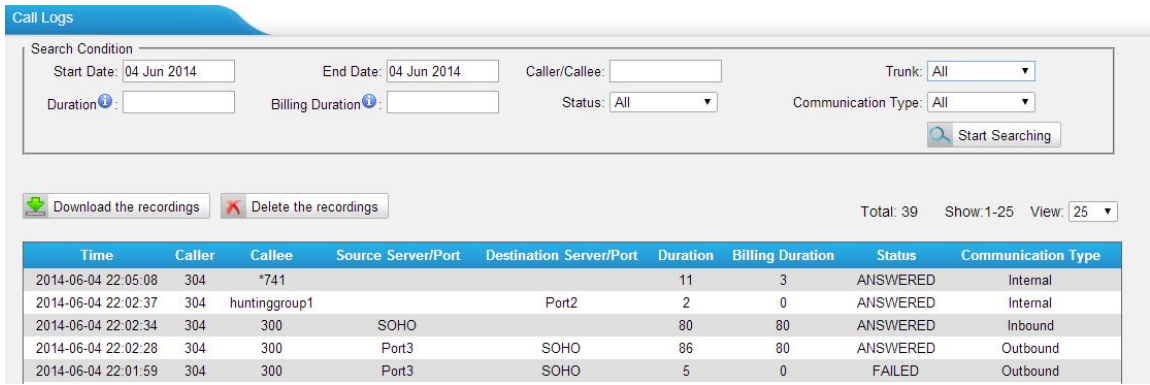


Figure 11-4 Call Logs

System Logs

You can download and delete the system logs of TA100/200.

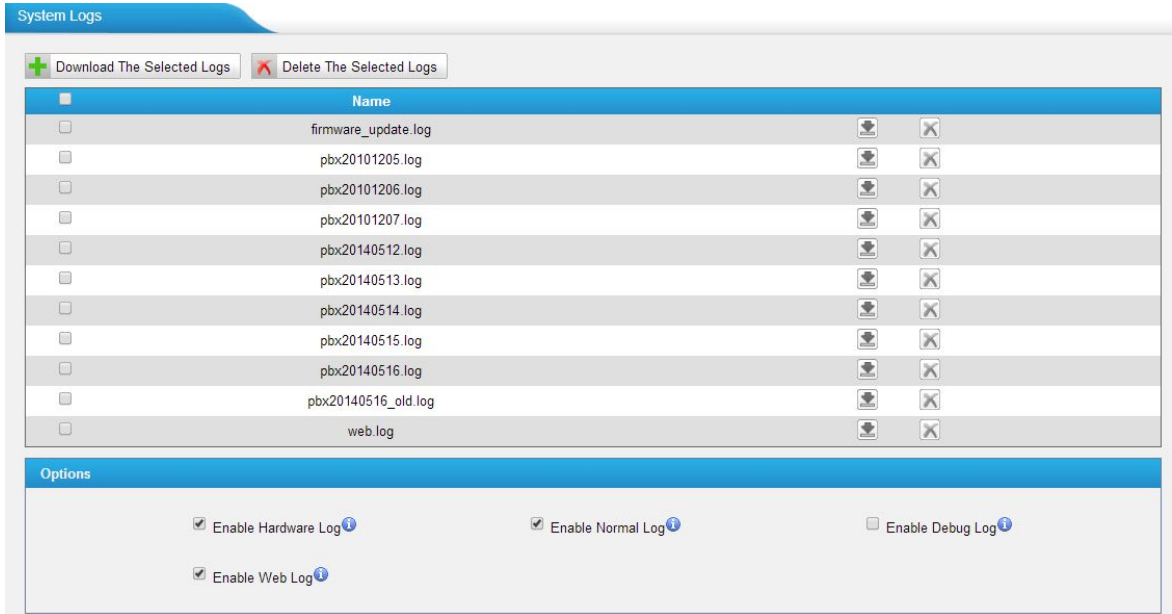


Figure 11-5 System Logs

- **Enable Hardware Log**
Save the information of hardware; (up to 4 log files)
- **Enable Normal Log**
Save the prompt information; (up to 16 log files)
- **Enable Web Log**
Save the history of web operations (up to 2 log files)
- **Enable Debug Log**
Save debug information (up to 2 log files)

Packet Tool

This feature is used to capture packets for technician. Integrate packet capture tool “Wireshark” in TA100/200.

Users also could specify the destination IP address and port to get the packets.

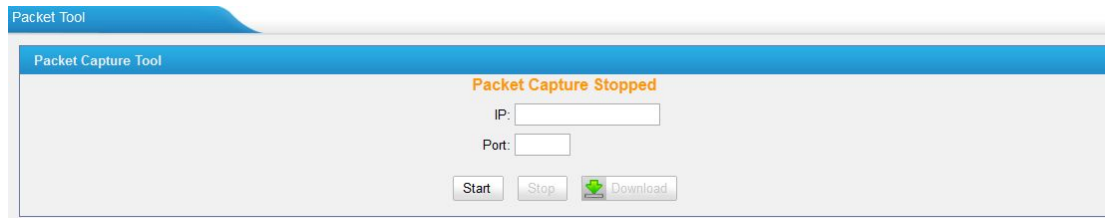


Figure 11-6 Packet Tool

- **IP**
Specify the destination IP address to get the packets.
- **Port**
Specify the destination Port to get the packets.

[END]